REMARKS/ARGUMENTS

This letter is responsive to the Office Action dated May 31, 2002. However, the Office Action was lost in the mail and received by the Applicant by fax on August 7, 2002. The USPTO granted a petition by the Applicant to have the response period reset to begin on August 7, 2002. Accordingly, this response is accompanied by a request for a one-month extension of time. The Applicant therefore submits that this response is timely filed.

By this response, claims 1-6, 8-10, 12-14, 17 and 21-24 have been amended and new claims 25-30 have been added. In addition, Figure 2 and various paragraphs in the description have also been amended. The Applicant submits that no new matter has been added by these claim amendments.

The Applicant has amended independent claims 1 and 21 to better claim the invention and has also revised independent claims 4 and 17 to depend from independent claim 1. Claim 1 now recites a method of reducing noise in an input signal that contains speech having a signal to noise ratio by incorporating the method steps recited in original claim 4 as filed and claiming that the input signal is modified by an attenuation function to generate a noise reduced signal wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios. Support for this amendment is on page 6, line 30 to page 9, line 2 of the application as originally filed.

Claim 21 now recites an apparatus for reducing noise in an input signal, the apparatus including an input for receiving the input signal and comprising a compression circuit for receiving a compression control signal and generating an amplification control signal in response, an amplification unit for receiving the input signal and the amplification control signal and generating an output signal with compression and reduced noise; and, an auxiliary noise reduction unit connected to the input for generating an auxiliary noise reduced signal where the compression control signal is the auxiliary noise reduced signal. Support for this amendment is on page 5, line 8 to page 5, line 31 as well as Figure 1 of the application as originally filed.

In light of the revision to claim 1, the Applicant has also amended claims 2 to 5 and added new claims 25 and 26 to better claim the invention. Claim 2 claims that the input signal is supplied to the amplification unit, the noise reduced signal is provided to the compression circuit for generating a control signal for the amplification unit and the amplification unit is controlled by the control signal to modify the input signal to generate an output signal with compression and reduced noise. Claim 3 now claims that providing the noise reduced signal to the compression circuit involves subjecting the input signal to an auxiliary noise reduction algorithm to generate the noise reduced signal. Claim 4 claims that the auxiliary noise reduction method claimed in claim 3 comprises the method of claim 1. Claim 5 has been amended to alternatively claim that the auxiliary noise reduction algorithm is different from the method of claim 1. Support for the amendments for claims 2 to 5 is in Figure 1 and lines 8 to 31 on page 5 of the application as originally filed.

Further, new claim 25 has been added to claim that the square of the speech magnitude spectral estimate is determined by subtracting the square of the noise magnitude spectral estimate from the square of the magnitude spectrum of the input signal. In addition, new claim 26 has been added to state that step (6) comprises applying the steps (1) to (5) to the input signal prior to supplying the input signal to the amplification unit. Support for these claim amendments are on page 7, lines 13 to 23 and on page 5, lines 26 to 31 and Figure 1, respectively, of the application as filed.

In light of the amendments to claim 1, claim 17 has been amended to depend from claim 1 and recite the features of the method used to detect the presence and absence of speech which is required in step (1) of claim 1.

In addition, in light of the amendments to claim 21, the Applicant has amended claims 22 and 23 to recite additional structural features for the apparatus and added new claims 27 to 30. New claim 27 claims that the apparatus alternatively comprises a main noise reduction unit connected to the input for generating a noise reduced signal that is supplied to the amplification unit in place of the input signal. New claim 28 claims that

the main noise reduction unit and the auxiliary noise reduction unit comprise a single unit. New claim 29 claims that, in the alternative, the auxiliary noise reduction unit is different from the main noise reduction unit. New claim 30 claims that the input signal has a signal to noise ratio and the noise filter calculation unit produces the noise reduced signal in dependence upon the signal to noise ratio, wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios. Support for claims 27 to 29 is on page 5, lines 8 to 31 of the application as filed and support for claim 30 is on page 6, line 30 to page 9, line 2 of the application as filed.

The Applicant has also amended claims 5, 6, 9, 12-14 and 22 to revise claim dependencies. In addition, the Applicant has amended claims 8, 9, 10, 22 and 24 to improve the readability of the claims and correct inadvertent errors. In particular, in the 4th line of claim 8, β has been replaced by $\hat{\beta}(f)$. In claims 9 and 10, the label (H(f)) has been added to each instance of the phrase "the attenuation function". Claim 22 has been amended to replace the phrase "detection means" with "detector", the phrase "a desired audio signal" with "speech" and the phrase "noise filter calculation means" with "noise filter calculation unit". Claim 24 has been amended to replace the phrase "noise filter calculation means" with "noise filter calculation unit" and to state that the noise filter calculation unit calculates an attenuation function (H(f)) rather than the auxiliary signal.

The description has also been amended so that the Summary of the Invention section corresponds with the claims as amended herein. The description has also been amended on page 6, line 5 of the application as filed to provide an application serial number and a publication number which were not available when the subject application was filed. In addition, line 3 on page 6 of the application as filed has been amended to specify that the proposed noise reduction technique is preferably carried out by noise reduction unit 18 (and possibly also noise reduction unit 16). Support for this amendment is in Figure 1 and page 6, lines 8 to 31 of the application as filed.

Figure 2 has also been amended to provide the reference numeral "46" at the output of the block identified by reference numeral "44" to better clarify the invention. The

description has also been amended on page 6, line 28 to comply with the change in Figure 2 by stating: "in known manner at 44, to provide a noise reduced signal 46. The noise reduced signal 46 in Figure 2 may correspond to either of the signals at 12 or 14 in Figure 1."

In the Office Action, the Examiner rejected claims 1 to 3 as being anticipated by Eguchi (U.S. Patent No, 5,337,366). The Examiner stated that Eguchi teaches a method for reducing noise in a signal comprising the steps of: 1) of supplying the input signal to an amplification unit, 2) subjecting the input signal to an auxiliary noise reduction algorithm to generate an auxiliary signal; 3) using the auxiliary signal to determine a control input for the amplification unit; and 4) controlling the amplification unit with a control signal to generate an output signal with reduced noise. The Examiner also stated that Eguchi teaches subjecting the input signal to a main noise reduction algorithm to generate a modified input signal which is supplied to the amplification unit and that the main and auxiliary noise reduction algorithms are different.

In response, the Applicant submits that amended claim 1 now recites a method of reducing noise in an input signal containing speech and having a signal to noise ratio. The method comprises: (1) detecting the presence and absence of speech; (2) in the absence of speech, determining a noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$; (3) in the presence of speech, comparing the magnitude spectrum of the input signal (|X(f)|) to the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$; (4) calculating an attenuation function ($H(f)$) from the magnitude spectrum of the input signal (|X(f)|) and the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$, the attenuation function ($H(f)$) being dependent on the signal to noise ratio; and, modifying the input signal by the attenuation function ($H(f)$) to generate a noise reduced signal wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios.

The Applicant submits that Eguchi does not teach the method steps of amended claim 1 nor does Eguchi teach the advantage of modifying an input signal by an attenuation

function wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios as taught by the Applicant's invention and claimed in amended claim 1. In addition, the Applicant submits that Euguchi does not teach the structural elements of the apparatus claimed in amended claim 21 nor does Eguchi teach the advantage of providing an auxiliary noise reduction unit for generating an auxiliary noise reduced signal that is provided to the compression circuit for generating an amplification control signal as taught in the Applicant's invention and claimed in amended claim 21.

The Applicant further submits that Eguchi teaches noise cancellation rather than noise reduction. In particular, Eguchi's system employs a feedback path to mitigate the effects of noise by producing a noise cancellation signal. In contrast, the Applicant's invention as claimed in amended claims 1 and 21, employs attenuation in the forward signal path to mitigate the effects of noise propagation. The Applicant submits that noise cancellation and noise reduction (i.e. attenuation) are quite different. In noise cancellation, sufficient knowledge is available about the noise signal to create a noise cancellation signal to cancel the noise signal when added to a signal that is corrupted by the noise signal. However, in the subject invention, the knowledge to create a noise cancellation signal is not available. Rather, statistical knowledge of the noise (in the form of a power spectrum) is used to create a noise filtering operation rather than a noise cancellation operation.

Accordingly, the Applicant respectfully submits that amended claims 1 and 21 are not anticipated or obvious in view of Eguchi and should be allowable. Furthermore, since claims 2-20 and 25-26 depend either directly or indirectly on claim 1, and since claims 22-24 and 27-30 depend either directly or indirectly on claim 21, the Applicant submits that claims 2-20 and 22-30 should also be allowable over Eguchi.

The Examiner also rejected claims 4 to 6 and 14 as being anticipated by Handel (PCT WO 96/241128). The Examiner stated that Handel teaches a method for reducing noise in an input audio signal containing speech, the method comprising: 1) detecting the presence and absence of speech utterances; 2) in the absence of speech, determining

a noise magnitude spectral estimate; 3) in the presence of speech comparing the magnitude of the audio signal to the noise magnitude spectral estimate; 4) calculating an attenuation function from the magnitude spectrum of the audio signal and the noise magnitude spectral estimate; and 5) modifying the input signal by the attenuation function to generate an output signal with reduced noise. The Examiner further stated that Handel determines the square of the speech magnitude spectral estimate by subtracting the square of the noise magnitude spectral estimate from the square of the magnitude spectrum of the input signal and that the attenuation function H(f) is calculated according to: $H(f) = \{(|X(f)|^2 - \beta|N(f)|^2)/|X(f)|^2)\}^{\alpha}$ where X(f) is the input signal magnitude spectrum, N(f) is the noise magnitude spectral estimate, β is an oversubtraction factor and α is an attenuation rule. The Examiner also stated that Handel further teaches a modified auto-correlation method (i.e. autoregressive on page 11, lines 14-15).

In response, the Applicant submits that Handel does not teach the use of a modified auto-correlation method to detect speech. Rather Handel teaches the use of an autoregressive model (AR) to model speech when speech is determined to be present. In particular, Handel discloses modeling speech by using an AR model (see page 11, line 20 to page 14, line 11). The Applicant submits that it is commonly known to those skilled in the art that one method of calculating the autoregressive method is to use the autocorrelation method in which the autocorrelation sequence of a signal is calculated and modeled by AR parameters. The Applicant submits that this is very different from using a modified autocorrelation function to detect speech as recited in claims 14 and 17 to 20 of the Applicant's patent application. In particular, the Applicant's modified autocorrelation function is directed towards improving computational efficiency by computing values of the autocorrelation sequence in parallel and not modeling the signal via an AR model as taught by Handel.

Furthermore, the Applicant submits that Handel does not teach the advantage of modifying an input signal by an attenuation function wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios as

taught by the Applicant's invention and claimed in amended claim 1. In addition, the Applicant submits that Handel does not teach the structural elements of the apparatus claimed in claim 21. In particular, Handel does not teach the advantage of providing an auxiliary noise reduction unit and a compression circuit in which the auxiliary noise reduction unit generates an auxiliary noise reduced signal for providing a compression control signal that is provided to the compression circuit which then generates an amplification control signal as taught in the Applicant's invention and claimed in amended claim 21. The advantage is that a cleaner signal is used to control the amplification unit.

Accordingly, the Applicant respectfully submits that amended claims 1 and 21 are not anticipated or obvious in view of Handel and should be allowable. Furthermore, since claims 2-20 and 25-26 depend either directly or indirectly from claim 1, and claims 22-24 and 27-30 depend either directly or indirectly on claim 21, the Applicant submits that claims 2-20 and 22-30 should also be allowable over Handel.

The Examiner also rejected claims 17 to 20, as being anticipated by Yasunaga (U.S. 4,845,753). The Examiner stated that Yasunaga teaches a method for determining the presence of speech in an audio signal comprising taking a block of an input audio signal and performing an auto-correlation on that block to form a correlated signal and checking the correlated signal for the presence of a periodic signal having a pitch corresponding to that for speech. The Examiner also stated that Yasunaga teaches that the autocorrelation is performed on a first block taken from an audio signal and a delayed block taken from the audio signal. The Examiner further stated that Yasunaga teaches that each block is subdivided into a plurality of shorter sections and the correlation comprises correlating pairs of the shorter sections to form partial correlations and subsequently summing the partial correlations to obtain the correlated signal. The Examiner further stated that Yasunaga teaches that the partial correlation involves storing the input signal as a plurality of samples in a pair of correlation buffers and that the autocorrelation is performed on the signals in the buffers to determine the partial correlations which are summed and stored.

In response, the Applicant submits that the Examiner's observations are incorrect since there is no mention of performing a correlation on a first block taken from an audio signal and a delayed block form the audio signal in col. 3, lines 45-58, nor is there any mention of subdividing each block into a plurality of shorter sections and performing correlations between pairs of the shorter sections to form partial correlations and subsequently summing the partial correlations to obtain the correlated signal. The Applicant also submits that there is no teaching of storing an input signal as a plurality of samples in a pair of correlation buffers and performing the correlation on the signals in the buffers to determine the partial correlations which are summed and stored.

The Applicant submits that Yasunaga is teaching an inverse filter approach for finding a pitch frequency. Accordingly, steps s41 through s50 represent the sequential calculations of the Durban recursion algorithm in which the prediction residual calculating circuit (3) and the order control circuit (4) calculate the PARCOR coefficients to model the input data according to the AR method (see col. 1, lines 11 to 40). As mentioned above, this operation is performed for modeling a signal and, as is well known to those skilled in the art, one modeling method is the autocorrelation method in which the autocorrelation sequence of a signal is first calculated and is then modeled by an AR model which calculates PARCORR coefficients.

In contrast to Yasunaga, in the subject invention, to detect whether speech is present in a computationally efficient manner, the Applicant teaches a method of segmentation for performing the auto-correlation of a segmented portions of a signal in parallel as claimed in claims 17 to 20. The Applicant submits that this is quite different from the autoregressive modeling taught by Yasunaga.

Furthermore, the Applicant submits that claims 17 to 20 have been amended to depend either directly or indirectly on amended claim 1 and that Yasunaga in no way teaches any of the elements of amended claim 1. Accordingly, the Applicant submits that claims 17 to 20 are not anticipated or obvious in view of Yasunaga and should be allowable.

The Examiner also rejected claims 21 to 23 as being anticipated by Borth (U.S. 4,628,529). The Examiner stated that Borth discloses an apparatus including an input signal, a noise reduced output signal and an auxiliary noise reduction means which receives the input signal and generates an auxiliary signal. The Examiner stated that the apparatus further includes an amplification means that is provided with the input signal and the auxiliary signal for control thereof to generate an output signal with reduced noise. The Examiner also stated that the auxiliary noise reduction means comprises a detection means connected to the input for providing a detection signal indicative of the presence of a desired audio signal, a magnitude means for determining an input signal magnitude spectrum; a spectral estimate means for generating a noise magnitude spectral estimate, and a noise filter calculation means for receiving the noise magnitude spectral estimate and the input signal magnitude spectrum and having an output for the auxiliary signal connected to the amplification means. The Examiner further stated that Borth also teaches that the auxiliary noise reduction means has a frequency transform means for transforming a signal into the frequency domain to provide a transformed signal wherein the magnitude means determines the magnitude spectrum from the transformed signal and the spectral estimate means determines the noise spectral estimate from the transformed signal.

In response, the Applicant submits that amended claim 21 now recites an apparatus having a compression circuit for receiving a compression control signal and generating an amplification control signal in response as previously discussed. The amplification unit receives the input signal and the amplification control signal and generates an output signal with compression and reduced noise. The advantage is that a cleaner signal is used to control the amplification unit rather than the original noisy input signal.

The Applicant further submits that Borth does not disclose a compression circuit or teach the advantage that a noise reduced signal can be used to generate an amplifier control signal to control an amplifier to generate a signal with compression and reduced noise. Furthermore, the Applicant submits that element 130 disclosed by Borth is not an amplification unit but rather a power spectrum modifier which performs spectral subtraction on an input signal containing speech and noise (see column 4, lines 21 to

45). Phase information is then added to the resulting signal which is then converted to the time domain and provided at speech signal output 158 (See Figure 1 of Borth).

Accordingly, the Applicant respectfully submits that amended claim 21 is not anticipated or obvious in view of Borth. Furthermore, since claims 22 to 23 depend either directly or indirectly from claim 21, the Applicant submits that claims 22 to 23 should be allowable over Borth.

The Examiner further rejected claims 12 and 13 as being obvious in view of Handel and the common general knowledge in which the Examiner stated that it is known to remotely turn an electronic device on and off. In light of this, the Examiner concluded that it would have been known to those skilled in the art and therefore obvious to automatically disable noise reduction in the presence of very light noise or extremely adverse environments which is used to protect the user from extremely loud environments.

In response, the Applicant submits that claims 12 and 13 are dependent on amended claim 1 which, as discussed above, is not obvious in view of Handel. Since, claims 12 and 13 incorporate the features claimed in amended claim 1, the Applicant submits that claims 12 and 13 should be allowable.

The Examiner also rejected claims 15 and 16 as being unpatentable over Handel in view of Nakajima (U.S. 4,283,601). In particular, the Examiner stated that Handel teaches a method of reducing noise which involves detecting speech with a modified auto-correlation that can be replaced by the partial auto-correlation function taught by Nakajima. The Examiner further stated that Nakajima teaches: 1) taking an input sample and separating it into short blocks and storing the blocks into correlation buffers, 2) correlating the blocks with one another to form partial correlations, and 3) summing the partial correlations to obtain a final correlation. In addition, the Examiner stated that both Handel and Nakajima teach a method of detecting speech by using the Fast Fourier Transform to generate partial correlations.

In response, the Applicant submits that Nakajima teaches the measurement of waveform information rather than the detection of speech (see col. 3, line 67 to col. 4, line 3). This is further supported in col. 11 which shows that the method is used for recognition of the person who generated the speech signal. Furthermore, Nakajima teaches that sampled data portions are subjected to spectral analysis to evaluate a predetermined parameter such as a partial auto-correlation coefficient and that these coefficients are applied to an inverse filter to set the characteristic thereof (see col. 6, line 17 to 27).

The Applicant further submits that the partial auto-correlation coefficients taught by Nakajima are used in a method to determine the coefficients of an AR model of the sampled data. In addition, the Applicant submits that the term "partial auto-correlation coefficient" relates to the sequential method in which the AR model is generated from the autocorrelation sequence of the sampled data. In fact, Nakajima states that the partial auto-correlation coefficient is evaluated by exploiting the well-known PAR-COR analyzing technique (see col. 6, lines 29-33) in which the autocorrelation coefficients of an AR model of order P+1 are related to the partial autocorrelation coefficient of the AR model of order P. Therefore, the autocorrelation sequence is already computed prior to the calculation of the PAR-COR coefficients.

In contrast, the Applicant's method of generating the auto-correlation function of a signal, as claimed in claim 15 and 16, is directed towards generating the autocorrelation function of an input signal in a computationally efficient manner and not towards AR modeling. In particular, the Applicant teaches using a segmentation method for determining modified autocorrelation coefficients from divided segments of the input signal to compute many coefficients in parallel. The Applicant submits that this is not similar to the recursive calculations taught by Nakajima in the generation of the PAR-COR coefficients.

In addition, the Applicant submits that claims 15 and 16 depend indirectly from amended claim 1 and therefore include the features recited in amended claim 1. As previously mentioned, amended claim 1 claims a method of reducing noise in an input

signal containing speech and having a signal to noise ratio by modifying the input signal by an attenuation function (H(f)) to generate a noise reduced signal wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios. The Applicant submits that Nakajima and Handel do not teach this feature.

Accordingly, in light of the arguments made above, the Applicant respectfully submits that amended claim 1, as well as claims 15 and 16, are not obvious in view of Nakajima and Handel and should be allowable.

The Examiner also rejected claim 24 as being unpatentable over Borth in view of Handel. The Examiner stated that Borth teaches an apparatus for reducing noise in a signal having a noise filter calculation means for determining the square of the speech magnitude spectral estimate by subtracting the square of the noise magnitude spectral estimate from the square of the input signal magnitude spectrum. The Examiner further stated that Handel teaches a method of reducing noise in an input audio signal containing speech wherein the attenuation function means calculates the auxiliary signal as an attenuation function with an oversubtraction factor and an attenuation rule as specified above. The Examiner concluded that it would have been obvious to combine Borth's system with the noise filter calculation means as calculated by Handel to calculate the auxiliary signal as an attenuation function with an oversubtraction factor and an attenuation rule.

In response, the Applicant respectfully submits that claim 24 depends indirectly on claim 21 and therefore includes all of the features recited in claim 21. As discussed previously, claim 21 is not obvious in light of the cited art. Accordingly, the Applicant submits that claim 24 is also not obvious in view of the cited art and is allowable.

The Examiner also stated that the Applicant should provide updated information regarding the patent application cited on page 6, line 5 of the specification. In response, the Applicant has amended the paragraph that begins on page 6, line 1 to indicate that the patent application cited on page 6, line 5 has an international application no. of PCT/CA98/00329 and an international publication of no. WO 98/47313.

Formal Drawings

The Applicant respectfully acknowledges the Notice of Draftsperson's Patent Drawing Review. Accordingly, the Applicant is providing formal drawings enclosed herein for review by the Drawing Review Branch.

Attached hereto is a marked-up version of the changes made to the specification and claims by the current amendment. The attached page is captioned: **Version with markings to show changes made.**

Conclusion

In view of the foregoing comments, it is respectfully submitted that the application is now in condition for allowance and the Applicant respectfully requests a timely Notice of Allowance be issued in this case. If the Examiner has any further concerns regarding the language of the claims or the applicability of the prior art, the Examiner is respectfully requested to contact the undersigned at 416-957-1687.

Respectfully submitted,

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TRO / cec

"VERSION WITH MARKINGS TO SHOW CHANGES MADE"

In the Drawings:

The applicant respectfully submits a marked up version for Figure 2 to indicate that the text "Noisy Signal" and "Enhanced Signal" has been removed and that the reference label "46" has been added at the output of the block labeled by the reference label "44" as shown in the Figures enclosed herein.

In the Description:

Please amend the paragraph beginning on page 1, line 28 as follows:

The present invention provides a two-fold approach to sound quality improvement under high noise situations and its practical implementation in a hearing aid. In one aspect, the ~~The~~ present invention removes noise from the input signal and controls a ~~the~~ compression stage with a cleaner signal, compared to the use of the original noisy input signal for controlling compression as is done in the prior art. The signal for amplification ~~(the upper path)~~ is, optionally, processed with a different noise reduction algorithm. Under certain circumstances, it may be desirable to use the same noise reduced signal for application and compression control in which case the two noise reduction blocks merge. In another instance, it may be desirable to alter or eliminate the noise reduction in the upper path.

Please amend the paragraph beginning on page 2, line 18 as follows:

In accordance with a first aspect of the present invention, there is provided a method of reducing noise in a signal containing speech and having a signal to noise ratio, the method comprising the steps:

(1) detecting the presence and absence of speech;

(2) in the absence of speech, determining a noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$;

(3) in the presence of speech, comparing the magnitude spectrum of the input signal (|X(f)|) to the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$;

(4) calculating an attenuation function (H(f)) from the magnitude spectrum of the input signal (|X(f)|) and the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$, the attenuation function (H(f)) being dependent on the signal to noise ratio; and,

(5) modifying the input signal by the attenuation function (H(f)), to generate a noise reduced signal wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios.

~~(1) supplying the input signal to an amplification unit;~~

~~(2) subjecting the input signal to an auxiliary noise reduction algorithm, to generate an auxiliary signal;~~

~~(3) using the auxiliary signal to determine control inputs for the amplification unit; and,~~

~~(4) controlling the amplification unit with the control signals, to generate an output signal with reduced noise.~~

Please insert the following paragraph before the paragraph beginning on page 2, line 28 as follows:

Preferably, the method further comprises the steps of:

(6) supplying the input signal to an amplification unit;

(7) providing the noise reduced signal to a compression circuit which generates a control signal for the amplification unit; and,

(8) controlling the amplification unit with the control signal to modify the input signal to generate an output signal with compression and reduced noise.

Advantageously, step (7) comprises subjecting the input signal to an auxiliary noise reduction algorithm to generate an auxiliary noise reduced signal and providing the auxiliary noise reduced signal to the compression circuit.

Please delete the paragraph beginning on page 2, line 28.

Please insert the following paragraphs before the paragraph beginning on page 2, line 32 as follows:

In one embodiment, step (6) comprises applying the steps (1) to (5) to the input signal prior to supplying the input signal to the amplification unit.

Furthermore, in one embodiment, the input signal may be subjected to a main noise reduction algorithm to generate a modified input signal which is supplied to the amplification unit. The auxiliary noise reduction algorithm may comprise the same noise reduction method as the main noise reduction algorithm. Alternatively, the auxiliary noise reduction algorithm may be different from the noise reduction method in the main noise reduction algorithm.

Please delete the paragraph beginning on page 2, line 32.

Please amend the paragraph beginning on page 3, line 12 as follows:

~~Preferably, the attenuation factor~~ Conveniently, the square of the speech magnitude spectral estimate $\left(\left|\hat{S}(f)\right|\right)$ may be determined by subtracting the square of the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ from the square of the magnitude spectrum of the input signal (|X(f)|). In a preferred embodiment, the attenuation factor is a function of frequency and is calculated in accordance with the following equation:

$$H(f) = \left[\frac{|X(f)|^2 - \beta\,|\hat{N}(f)|^2}{|X(f)|^2}\right]^{\alpha}$$

where f denotes frequency, $H(f)$ is the attenuation function, $|X(f)|$ is the magnitude spectrum of the input audio signal; $\left(\left|\hat{N}(f)\right|\right)$ is the noise magnitude spectral estimate, β is an oversubtraction factor and α is an attenuation rule, wherein α and β are selected to give a desired attenuation function. The oversubtraction factor β is, preferably, varied

as a function of the signal to noise ratio, with β being zero for high and low signal to noise ratios and with β being increased as the signal to noise ratio increases above zero to a maximum value at a predetermined signal to noise ratio and for higher signal to noise ratios β decreases to zero at a second predetermined signal to noise ratio greater than the first predetermined signal to noise ratio.

Please amend the paragraph beginning on page 3, line 25 as follows:

Advantageously, the oversubtraction factor β is divided by a preemphasis function of frequency P(f) to give a modified oversubtraction factor $\hat{\beta}(f)$, the preemphasis function being such as to reduce $\hat{\beta}(f)$ ~~β~~ at high frequencies, to reduce attenuation at high frequencies.

Please amend the paragraph beginning on page 4, line 12 as follows:

Another aspect of the present invention provides for a method of detecting ~~determining~~ the presence or the absence of speech in an audio signal, the method comprising taking a block of ~~an input~~ the audio signal and performing an auto-correlation on that block to form a correlated signal; and checking the correlated signal for the presence of a periodic signal having a pitch corresponding to that for speech.

Please insert the following paragraphs after the paragraph beginning on page 4, line 18 as follows:

In a further aspect the present invention provides an apparatus for reducing noise in an input signal, the apparatus including an input for receiving the input signal. The apparatus comprises a compression circuit for receiving a compression control signal and generating an amplification control signal in response, and an amplification unit for receiving the input signal and the amplification control signal and generating an output signal with compression and reduced noise. The apparatus further comprises an

auxiliary noise reduction unit connected to the input for generating an auxiliary noise reduced signal, the compression control signal being the auxiliary noise reduced signal.

The apparatus may further comprise a main noise reduction unit connected to the input for generating a noise reduced signal and supplying the noise reduced signal in place of the input signal to the amplification unit.

Preferably, the input signal contains speech and the main noise reduction unit comprises a detector connected to the input and providing a detection signal indicative of the presence of speech and a magnitude means for determining the magnitude spectrum of the input signal (|X(f)|), with both the detector and the magnitude means being connected to the input of the apparatus. The main noise reduction unit further comprises a spectral estimate means for generating a noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ and being connected to the detector and to the input of the apparatus, a noise filter calculation unit connected to the spectral estimate means and the magnitude means, for receiving the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ and magnitude spectrum of the input signal (|X(f)|) and calculating an attenuation function $(H(f))$, and a multiplication unit coupled to the noise filter calculation unit and the input signal for producing the noise reduced-signal.

Please delete the paragraph beginning on page 4, line 18.

Please amend the paragraph beginning on page 5, line 1 as follows:

Figure 1 is a conceptual blocked diagram for hearing aid noise reduction and compression;

Please amend the paragraph beginning on page 5, line 16 as follows:

Here, the position of the noise reduction unit 18 can advantageously provide~~s~~ a cleaner signal for controlling the compression stage. The noise reduction unit 18 provides a first generating means which generates an auxiliary signal from an auxiliary

noise reduction algorithm. The auxiliary algorithm performed by unit 18 may be identical to the one performed by unit 16, except with different parameters. Since the auxiliary noise reduced signal is not heard, unit 18 can reduce noise with increased aggression. This auxiliary signal, in turn, controls the compression circuitry 20, which comprises second generating means for generating a control input for controlling the amplification unit 22.

Please amend the paragraph beginning on page 6, line 1 as follows:

With reference to Figure 2, this shows a block diagram of a ~~hearing aid with a~~ specific realization of the proposed noise reduction technique which is preferably carried out by noise reduction unit 18 (and possibly also noise reduction unit 16). The incoming signal at 10 is first blocked and windowed, as detailed in applicant's simultaneously filed international application serial no. PCT/CA98/00329 corresponding to international publication no. WO 98/47313 which is incorporated herein by reference. The blocked and windowed output provides the input to the frequency transform (all of these steps take place, as indicated, at 32), which preferably here is a Discrete Fourier Transform (DFT), to provide a signal X(f). The present invention is not however restricted to a DFT and other transforms can be used. A known, fast way of implementing a DFT with mild restrictions on the transform size is the Fast Fourier Transform (FFT). The input 10 is also connected to a speech detector 34 which works in parallel to isolate the pauses in the incoming speech. For simplicity, reference is made here to "speech", but it will be understood that this encompasses any desired audio signal, capable of being isolated or detected by detector 34 ~~including music~~. These pauses provide opportunities to update the noise spectral estimate. This estimate is updated only during speech pauses as a running slow average. When speech is detected, the noise estimate is frozen.

Please amend the paragraph beginning on page 6, line 25 as follows:

A noise filter calculation 40 is made based on |X(f)| and $|\hat{N}(f)|$, to calculate an attenuation function H(f). As indicated at 42, this is used to control the original ~~input~~

noisy signal X(f) by multiplying X(f) by H(f). This signal is subject to an inverse transform and overlap-add resynthesis in known manner at 44, to ~~give an output at 44.~~ provide a noise reduced signal 46. The noise reduced signal 46 in Figure 2 may correspond to either of the signals at 12 or 14 in Figure 1.

In the Claims:

Please amend claims 1-6, 8-10, 12-14, 17 and 21-24 as shown below. Furthermore, please add new claims 25-30 as shown below.

1. (Amended) A method of reducing noise in ~~a~~ an input signal, said input signal containing speech and having a signal to noise ratio, the method comprising the steps:

~~(1) supplying the input signal to an amplification unit;~~

~~(2) subjecting the input signal to an auxiliary noise reduction algorithm, to generate an auxiliary signal;~~

~~(3) using the auxiliary signal to determine a control input for the amplification unit; and~~

~~(4) controlling the amplification unit with the control signal, to generate an output signal with reduced noise.~~

(1) detecting the presence and absence of speech;

(2) in the absence of speech, determining a noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$;

(3) in the presence of speech, comparing the magnitude spectrum of the input signal (|X(f)|) to the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$;

(4) calculating an attenuation function (H(f)) from the magnitude spectrum of the input signal (|X(f)|) and the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$, the attenuation function (H(f)) being dependent on the signal to noise ratio; and,

(5) modifying the input signal by the attenuation function (H(f)) to generate a noise reduced signal wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios.

2. (Amended) A method as claimed in claim 1, further comprising the steps of: ~~wherein the input signal is subjected to a main noise reduction algorithm, to generate a modified input signal, which is supplied to the amplification unit.~~

(6) supplying the input signal to an amplification unit;

(7) providing the noise reduced signal to a compression circuit which generates a control signal for the amplification unit; and

(8) controlling the amplification unit with the control signal to modify the input signal to generate an output signal with compression and reduced noise.

3. (Amended) A method as claimed in claim 2, wherein ~~the main and auxiliary noise reduction algorithms are different.~~ step (7) comprises subjecting the input signal to an auxiliary noise reduction algorithm to generate an auxiliary noise reduced signal and providing the auxiliary noise reduced signal to the compression circuit.

4. (Amended) A method as claimed in claim 3, wherein the auxiliary noise reduction algorithm comprises the method of claim 1.~~of reducing noise in an input, audio signal containing speech, the method comprising the steps of:~~

~~(a) detecting the presence and absence of speech utterances;~~

~~(b) in the absence of speech, determining a noise magnitude spectral estimate;~~

~~(d) in the presence of speech, comparing the magnitude spectrum of the audio signal to the noise magnitude spectral estimate;~~

~~(e) calculating an attenuation function from the magnitude spectrum of the audio signal and the noise magnitude spectral estimate; and~~

~~(f) modifying the input signal by the attenuation function, to generate an output signal with reduced noise.~~

5. (Amended) A method as claimed in claim ~~4~~3, wherein the auxiliary noise reduction algorithm is different from the method of claim 1.~~square of the speech magnitude spectral estimate is determined by subtracting the square of the of the noise magnitude spectral estimate from the square of the magnitude spectrum of the input signal.~~

6. (Amended) A method as claimed in claim ~~5~~25, wherein the attenuation function is calculated in accordance with the following equation:

$$H(f) = \left[\frac{|X(f)|^2 - \beta |\hat{N}(f)|^2}{|X(f)|^2} \right]^\alpha$$

where H(f) is the attenuation function, |X(f)| is the magnitude spectrum of the input signal; $|\hat{N}(f)|$ is the noise magnitude spectral estimate, β is an oversubtraction factor and α is an attenuation rule, wherein α and β are selected to give a desired attenuation function.

8. (Amended) A method as claimed in claim 7, wherein the oversubtraction factor β is divided by a preemphasis function P(f) to give a modified oversubtraction factor $\hat{\beta}(f)$, the preemphasis function being such as to reduce ~~β~~ $\hat{\beta}(f)$ at high frequencies, and thereby reduce attenuation at high frequencies.

9. (Amended) A method as claimed in claim 6, ~~7 or 8,~~ wherein the rate of change of the attenuation function (H(f)) is controlled to prevent abrupt and rapid changes in the attenuation function (H(f)).

10. (Amended) A method as claimed in claim 6, wherein -the attenuation function (H(f)) is calculated at successive time frames, and the attenuation function (H(f)) is calculated in accordance with the following equation:

$$G_n(f) = (1-\gamma)H(f) + \gamma G_{n-1}(f)$$

wherein $G_n(f)$ and $G_{n-1}(f)$ are the smoothed attenuation functions at the n'th and (n-1) 'th time frames, and γ is a forgetting factor.

12. (Amended) A method as claimed in claim ~~4~~1 which includes remotely turning noise suppression on and off.

13. (Amended) A method as claimed in claim ~~4~~1 which includes automatically disabling noise reduction in the presence of very light noise or extremely adverse environments.

14. (Amended) A method as claimed in claim ~~4~~1 which includes detecting speech with a modified auto-correlation function.

17. (Amended) A method as claimed in claim 1, wherein detecting ~~of determining~~ the presence or absence of speech ~~in an audio signal, the method comprising~~ comprises:

(1) taking a block of ~~an~~ the input ~~audio~~ signal and performing an auto-correlation on that block to form a correlated signal; and,

(2) checking the correlated signal for the presence of a periodic signal having a pitch corresponding to that for ~~the~~ a desired audio signal.

21. (Amended) An apparatus, for reducing noise in ~~a~~ an input signal, the apparatus including an input for receiving the ~~a~~ input signal ~~and an output for a noise reduced signal~~, the apparatus comprising:

(a) a compression circuit for receiving a compression control signal and generating an amplification control signal in response;

(b) an amplification unit for receiving the input signal and the amplification control signal and generating an output signal with compression and reduced noise; and,

(c) an auxiliary noise reduction unit connected to the input for generating an auxiliary noise reduced signal, the compression control signal being the auxiliary noise reduced signal.

~~(a) an auxiliary noise reduction means connected to the input for generating an auxiliary signal; and~~

~~(b) an amplification means connected to the input for receiving the original input signal and to the auxiliary noise reduction means, for receiving the auxiliary signal, the amplification means being controlled by the auxiliary signal to generate an output signal with reduced noise.~~

22. (Amended) An apparatus as claimed in claim ~~21~~27, wherein the input signal contains speech and the main ~~auxiliary~~ noise reduction unit ~~means~~ comprises:

(1) a detector ~~detection means~~ connected to said input and providing a detection signal indicative of the presence of speech ~~a desired audio signal~~;

(2) magnitude means for determining the magnitude spectrum of the input signal (|X(f)|), with both the detector ~~detection means~~ and the magnitude means being connected to the input of the apparatus;

(3) spectral estimate means for generating a noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ and being connected to the detector ~~detection means~~ and to the input of the apparatus; ~~and~~

(4) a noise filter calculation ~~means~~ unit connected to the spectral estimate means and the magnitude means, for receiving the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ and magnitude spectrum of the input signal (|X(f)|) and calculating an attenuation function (H(f)); and, ~~to produce the auxiliary signal and having an output for the auxiliary signal connected to the amplification means.~~

(5) a multiplication unit coupled to the noise filter calculation unit and the input signal for producing the noise reduced signal.

23. (Amended) An apparatus as claimed in claim 22, which includes a frequency transform means connected between said input and both of the magnitude means and the spectral estimate means for transforming the signal into the frequency domain to provide a transformed signal (X(f)) wherein the magnitude means determines the magnitude spectrum (|X(f)|) from the transformed signal (X(f)), and wherein the spectral estimate means determines the noise spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ from the transformed signal (X(f)) in the absence of speech, the apparatus further including inverse frequency transform means for receiving a transformed noise reduced signal from the multiplication unit, the inverse frequency transform means providing the noise reduced signal.

24 (Amended) An apparatus as claimed in claim 23, wherein the noise filter calculation ~~means~~ unit determines the square of the speech magnitude spectral estimate by subtracting the square of the noise magnitude spectral estimate from the square of the magnitude spectrum of the input signal and wherein the noise filter calculation ~~means~~ unit calculates the ~~auxiliary signal as an~~ attenuation function (H(f)), as a function of frequency, in accordance with the following equation:

$$H(f) = \left[\frac{|X(f)|^2 - \beta |\hat{N}(f)|^2}{|X(f)|^2} \right]^{\alpha}$$

where f denotes frequency, $H(f)$ is the attenuation function, $|X(f)|$ is the magnitude spectrum of the input audio signal; $|\hat{N}(f)|$ is the noise magnitude spectral estimate, β is an oversubtraction factor and α is an attenuation rule, wherein α and β are selected to give a desired attenuation function.

25. (New) A method as claimed in claim 1, wherein the square of the speech magnitude spectral estimate $(|\hat{S}(f)|)$ is determined by subtracting the square of the noise magnitude spectral estimate $(|\hat{N}(f)|)$ from the square of the magnitude spectrum of the input signal (|X(f)|).

26. (New) A method as claimed in claim 2, wherein step (6) comprises applying steps (1) to (5) to the input signal prior to supplying the input signal to the amplification unit.

27. (New) An apparatus as claimed in claim 21, wherein the apparatus further comprises a main noise reduction unit connected to the input for generating a noise reduced signal and supplying the noise reduced signal to the amplification unit in place of the input signal.

28. (New) An apparatus as claimed in claim 27, wherein the main noise reduction unit and the auxiliary noise reduction unit comprise a single unit.

29. (New) An apparatus as claimed in claim 27, wherein the auxiliary noise reduction unit is different from the main noise reduction unit.

30. (New) An apparatus as claimed in claim 22, wherein the input signal has a signal to noise ratio and the noise filter calculation unit produces the noise reduced signal in dependence upon the signal to noise ratio, wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios.